--The invention relates to a decoder for decoding an encoded digital signal that has been obtained by encoding a wideband digital signal of a specific sampling frequency F_s, for example a digital audio signal, in an encoder; and more particularly to a decoder for such an encoded digital signal comprising consecutive frames, where each frame comprises a plurality of information packets, each information packet comprising N bits, N being larger than 1. A frame comprises at least a first frame portion including synchronization information. The decoder has an input for receiving the encoded digital signal, ~~the decoder~~ and is adapted to convert the encoded digital signal into a replica of the wideband digital signal. The decoder has an output to supply the replica of the wideband digital signal. The invention also relates to a receiver comprising such a decoder. --

Pages 3, 4 and 5 Delete in their entirety and substitute:

--SUMMARY OF THE INVENTION

An object of the invention is to provide an encoder which can economically decode a digital signal having frames divided into differing numbers of packets.

Another object of the invention is to decode a digital signal so as to obtain a replica of the wideband signal.

According to the invention, the decoder provides a faithful replica of the original wideband signal when the number B of information packets in one frame has a relation to a value P, such that, if P in the formula

$$P = \frac{BR}{N} \times \frac{n_s}{F_s}$$

is an integer, where BR is the bitrate of the encoded digital signal and n_s is the number of samples of the wideband digital signal whose corresponding information in the encoded digital signal is included in one frame of the encoded digital signal, the number B of information packets in one frame is P; and if P is not an integer, the number B of information packets in a number of frames is P', where P' is the next lower integer ~~following~~ below P, and the number of information packets in the other frames is equal to P'+1 so as to exactly comply with the requirement that the average framerate of the encoded digital signal is substantially equal to F_s/n_s.

The purpose of dividing the frames into B information packets is that, for a wide-band digital signal of a sampling frequency F_s, the average frame rate of the encoded digital signal received is now such that the duration of a frame in the digital signal corresponds to the duration occupied by n_s

samples of the wide-band signal.

Preferably, the first frame portion further contains information related to the number of information packets in a frame. In a frame comprising B information packets this information may be equal to the value B. This means that this information corresponds to P' for frames comprising P' information packets and to P'+1 for frames comprising P'+1 information packets. Another possibility is that this information corresponds to P' for all frames, regardless of whether a frame comprises P' or P'+1 information packets. The additionally inserted (P'+1)th information packet may comprise for example merely "zeros". In that case this information packet does not contain any useful information. Of course, the additional information packet may also be filled with useful information.

The first frame portion may further comprise system information. This may include the sample frequency F_s of the wide-band digital signal applied to the transmitter, copy-protection codes, the type of wide-band digital signal applied to the transmitter, such as a stereo-audio signal or a mono-audio signal, or a digital signal comprising two substantially independent audio signals. However, other system information is also possible, such as the bitrate, as will become apparent hereinafter. Including the system information makes it possible for the receiver and thus the decoder in the receiver to be flexible and enables the received digital signal to be correctly reconverted into the wide-band digital signal.

The frames may comprise second and the third frame portions. Those frame portions contain other signal information, such as allocation information, quantized samples

and scale factor information. Upon encoding, the wideband digital signal can be split up so as to generate a number of M subsignals, M being larger than 1. Those subsignals are quantized so as to obtain quantized subsignals. For this purpose an arbitrary encoding, such as a transform coding or a subband coding, may be used. At the receiving end it is then necessary to apply an inverse encoding to recover the wideband digital signal.

In order to make the signal information available for decoding, the decoder is provided with retrieval means for retrieving the allocation information, the quantized samples and the scale factor information.

Preferably, the allocation information is inserted in a frame before the samples. This allocation information is needed to enable the continuous serial bit stream of the samples in the third frame portion to be subdivided into the various individual quantized samples of the correct number of bits at the receiving end. An adaptive bit allocation are described <u>inter alia</u> in the publication "Low bit-rate coding of high quality audio signals. An introduction to the MASCAM system" by G. Theile et al, EBU Technical Review, No. 230 (August 1988). Inserting the allocation information in a frame before the samples in a frame has the advantage that in the decoder a simpler decoding becomes possible, which can be carried out in real time and which produces only a slight signal delay. As a result of this sequence it is no longer necessary to first store all the information in the third frame portion in a memory in the decoder. Upon arrival of the encoded digital signal the allocation information is stored in a memory in the decoder. Information content of the allocation

information is much smaller than the information content of the samples in the third frame portion, so that a substantially smaller store capacity is needed than in the case that all the samples would have to be stored in the decoder. Immediately upon arrival of the serial data stream of the quantized samples in the third frame portion this data stream can be divided into the various samples having the number of bits specified by the allocation information, so that no previous storage of the signal information is necessary.

The allocation information may be in the form of 4-bit words and the scale factor information may be in the form of 6-bit words. The scale factor information is also inserted in the third frame portion before the samples, so that it is possible that during reception the scale factors derived from said scale information are first stored in a memory and the samples are multiplied immediately upon arrival, i.e. without a time delay, by the values of said scale factors.

B3

Moreover, it is evident that if after quantisation in the transmitter the subband signal in a subband is zero, which obviously will be apparent from the allocation information for the subband, no scale factor information for this subband need to be transmitted.

The inventive steps may be applied to decoders to be used in digital transmission systems, for example systems for the transmission of digital audio signals (digital audio broadcast) via the ether. However, other uses are also conceivable. An example of this is a transmission via optical or magnetic media. Optical-media transmissions may be, for example, transmissions via glass fibres or by means of optical discs or tapes. Magnetic-media transmissions are

possible, for example, by means of a magnetic disc or a magnetic tape. The encoded digital signal is then stored in the format as described in one or more tracks of a record carrier, such as an optical or magnetic disc or a magnetic tape.

The versatility and flexibility of the decoder thus resides in the special format with which the information in the form of the encoded digital signal is transmitted, for example via a record carrier. The decoder extracts said system information from the data stream and employs it for a correct decoding.

B3

The information packets constitute a kind of fictitious units, which are used to define the length of a frame. This means that they need not be explicitly discernible in the information stream of the encoded digital signal. Moreover, the relationship of the information packets with the existing digital audio interface standard is as defined in the IEC standard no. 958. This standard as normally applied to consumer products defines frames containing one sample of both the left-hand and the right-hand channel of a stereo signal. These samples are represented by means of 16-bit two's complement words. If N = 32 is selected, one frame of this digital audio interface standard can transmit exactly one information packet of the second digital signal. In the digital audio interface standard the frame rate is equal to the sample rate. For the present purpose the frame rate should be selected to be equal to BR/N. This enables the present ICs employed in standard digital audio interface equipment to be used.--